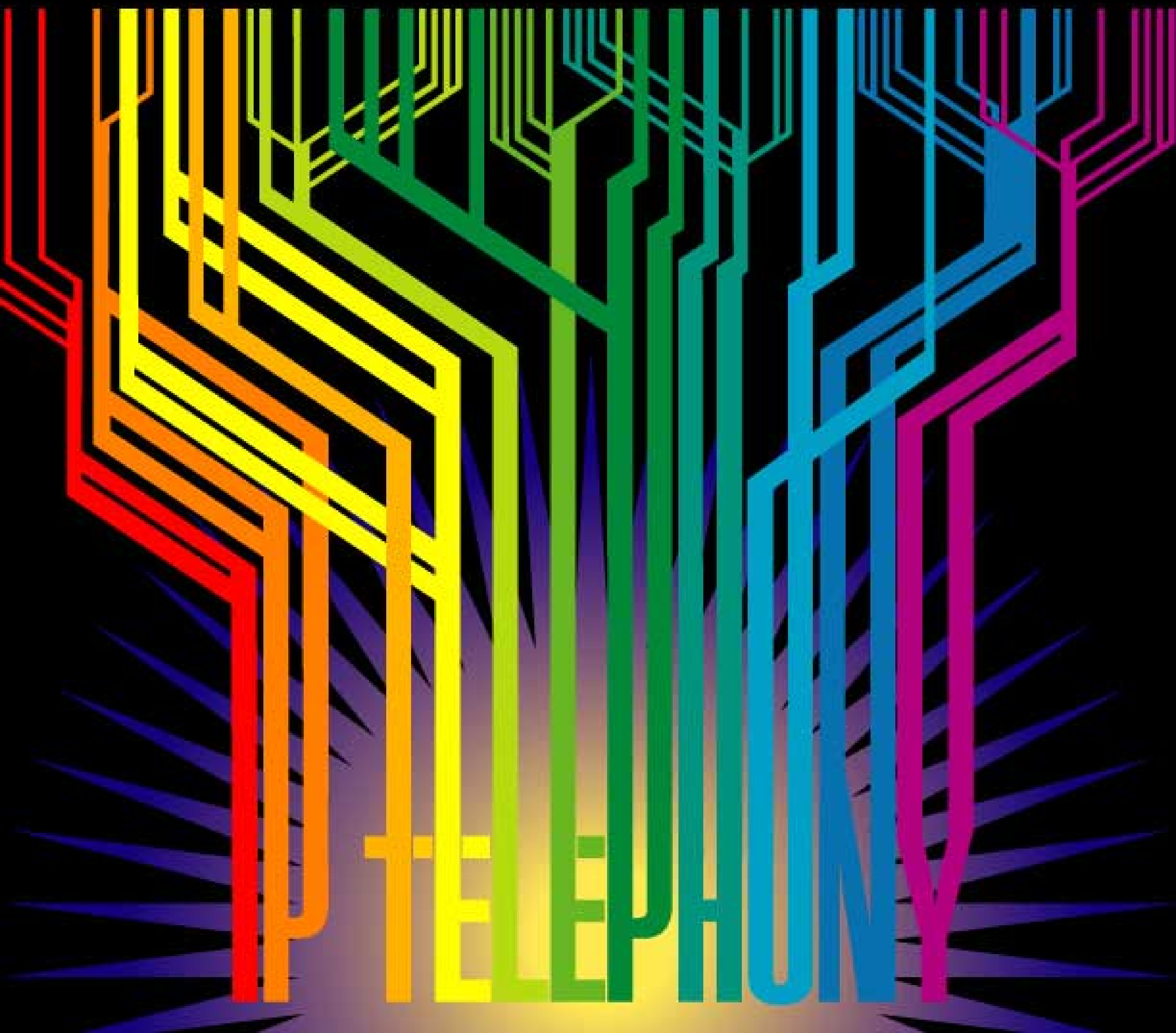
A SUPPLEMENT TO BUSINESS COMMUNICATIONS REVIEW

GETTING READY FOR IP TELEPHONY



Introduction

The Internet provides enormous opportunities to improve how we communicate, how we relate to one another, even how we live. It offers the potential for lowering costs, increasing efficiency and enhancing business-to-customer relationships, and the Internet is at the heart of the move toward the unification of disparate communication networks.

At Nortel Networks, we recognize the changes embodied by the Internet, and are helping to lead the way. But, much end-user uncertainty still exists. With all the noise about the Internet, voice over IP, IP Telephony and the like, customers understandably are confused. How much IP do they need? Where in their operations do they need IP? When do they need it? For that matter, do they need it at all? Will they have to throw out all their existing systems and start fresh? Is the PBX dead? Will voice be free?

The dynamism of the industry is such that there are few clear, comprehensive answers. One thing is certain, though—the influence of IP technology will continue to increase, and businesses are well advised to learn about it and to be prepared for it.

With this supplement to *Business Communications Review*, Nortel Networks hopes to give readers a place to begin finding answers. Working with *BCR*, we have designed the supplement to provide solid information to help network decision-makers devise and execute migration plans.

The supplement begins with an overview of how IP telephony is evolving. It includes discussions of how the PBX is integrating IP capability and of emerging solutions based on the IP networking model. The article also highlights some major issues and significant opportunities with IP Telephony.

Next, the supplement moves into a description of products and technology that are available now and a discussion of what is on the horizon. This analysis includes strengths and weaknesses of each so users can begin to evaluate the situation and formulate opinions.

And just how does one build a business case for IP Telephony? How does one go about deciding what is needed, how much is needed and where it is needed? The next article will provide some answers.

Finally, the supplement includes write-ups about two different types of organizations that have chosen to implement IP in different ways. Their representatives talk about why they made the decisions they made, and the results.

All the articles highlight a key point: The communication technology landscape is all about expanding choices. Enterprises can integrate new capabilities when the time is right and in ways that work best for them, and choose the technologies and ways of communicating that are most appropriate for their situations.

The Internet is changing everything. I hope this supplement will help you make the most of these new opportunities.

Sincerely,

Rick Moran
Vice President, Worldwide Marketing
Enterprise Solutions
Nortel Networks

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Combining Telephony and IP for Business Advantage

Tony Rybczynski

The Internet and the growing convergence around IP present great opportunities for businesses to capture new markets, serve customers better, reduce costs and improve productivity. However, these developments have yet to impact one key area: direct, person-to-person, real-time communication. This manifests itself in various ways, such as the absence of the human touch in e-commerce applications and the lack of tools that enhance distributed collaboration.

To address that gap, CIOs are analyzing business cases that bring together the best of two worlds: feature- and application-rich telephony environments, and content/transaction-rich intranet and Internet environments. This article explores the key enabling technologies that can help that effort —with a major emphasis on IP Telephony—and strategies enterprises can use to begin migrating toward convergence.

Technology Enablers

During the past two decades, telephone systems steadily evolved as businesses needed reliable communication that could overcome the barriers of time and distance. What were once basic voice communication systems have matured into sophisticated, enterprise-wide, multimedia communications platforms, that deliver a broad range of telephony services. The networking services available on these platforms include automatic least-cost routing and class-of-service routing, and applications such as voice mail, mobility and call centers.

During that same two-decade period, packet switching also matured, driven by applications such as file transfer, transaction processing and information access. First implemented as proprietary systems running over private lines, packet switching has evolved into standards-based, virtual-circuit networks (frame relay, for example) and the Internet. The arrival of Ethernet in the early 1980s led to bridges and routers and, more recently, LAN switches. Speeds have increased, prices have come down and there are now more than 200 million Internet and Ethernet users worldwide.

The worlds of circuit-switched voice and packet-switched data evolved separately, although there was some application integration via computer-telephony integration (CTI). Though the industry acknowledged the long-term trend toward convergence, it took some fundamental advances in technology before it could be considered practical. These included:

- Digital signaling processing (DSP), that can provide toll-quality, variable-bit-rate voice at considerably less than 64 kbps, and high-quality video at megabit speeds.
- The emergence of multimedia PCs, and audio- and video-streaming applications.
- The proven economics of transporting voice over packet networks, initially over variable bit rate-realtime (VBR-rt), ATM-switched virtual circuits (SVCs) with their rich Quality-of-Service (QOS) capabilities and, more recently, over QOS-enriched frame relay SVCs.
- IP QOS standards, queuing and traffic-management technologies, which take IP networking beyond the domain of "best-effort" transport of data packets.
- The rapid penetration of ecommerce over the Internet, and the need for the human touch that enables better customer service and differentiated services.

Two other major developments enabling convergence are the *Optical Internet* and the Wireless Internet. The Optical Internet, together with the application of high-capacity optical transmission and switching in enterprise wide-area and campus networks, promises to help network managers get ahead of the bandwidth demand curve. The *Wireless Internet* will bring the hundreds of millions of wireless users onto the Internet through wide-band, packet-radio technologies that support telephony, data and video, and transform the network from a bottleneck into a facilitator for application development.

All these technological developments combine to put telephony at the center of the migration to convergence. IP Telephony will rely on a switched network infrastructure to achieve toll-quality voice communication. In addition, it will include the intracorporate and public voice-numbering plans so crucial to today's enterprise networks.

IP Telephony also will have to deliver the kinds of interpersonal communication capabilities to which end users have grown accustomed. These include calling-line ID, three-way calling and voice mail. These capabilities are accessible to many end-user devices—including analog phones, featured telephones, ISDN phones, centralized attendant consoles, payphones and wireless handsets—through a relatively intuitive user interface (UI): the familiar dial pad and feature keys.

IP Telephony and Service Providers

Much of the interest in IP Telephony has been spurred by developments in Internet telephony as an alternative to the long-distance, Public Switched Telephone Network (PSTN). Datamonitor forecasts that by 2002, IP Telephony will account for 3% of U.S. long-distance voice traffic and as much as 14% of international phone traffic.

A number of service providers already have announced IP Telephony services, including AT&T, IDT, ITXC, Japan Telecom, PSINet, Qwest, DeltaThree, Sprint and TCI. These generally provide lower-price/lower-quality offerings and target general consumers, not business customers.

These carriers meet voice performance requirements by provisioning bandwidth for low packet loss and less-than-100-msec latency. Delta Three, however, has taken a different approach: It established what it claims is the world's largest dedicated overlay network for telephony and fax traffic. Techniques of provisioning additional bandwidth or building separate overlay networks for IP Telephony are not financially viable in the enterprise market, hence the need for QOS and sophisticated traffic management capabilities.

Delivering High Transmission Quality Using Standards

The biggest challenge facing IP Telephony will be accommodating business-critical applications. These include call centers, Interactive Voice Response (IVR) and other speech-activated applications, mobility and single-number roaming services, and unified messaging.

These applications put the spotlight on the need for IP Telephony to address the tough issue of transmission quality. Over the decades of telephone network operation, the network has become remarkably reliable and delivers consistently high-quality service. All of us have probably experienced noisy lines and excessive delay, but the instances are noteworthy precisely because they're so rare.

By contrast, on today's intranets and the public Internet, QOS is virtually nonexistent. You can never be sure how long it will take to download a file or pull up a web site, nor how long it will take email to reach its intended destination.

Throwing bandwidth at this problem is a short-term, partial fix; for the long term, other strategies are required. That is why the industry has embarked on several initiatives to establish standards for IP QOS, which would apply to both multimedia and business-critical data applications. These include:

- **Integrated Services (Int-Serv) Architecture**: Int-Serv includes the specification of the Resource reSerVation Protocol (RSVP). RSVP enables an application to specify what level of network resources it needs—bandwidth and performance—and enables routers to allocate resources accordingly. The Int-Serv approach, however, is limited in scalability, because each router along a path must maintain and manage information for each application flow that crosses the network.
- **Differentiated Services (Diff-Serv):** A simpler framework than Int-Serv, it does not specify per-flow-state signaling, which significantly enhances its scalability. Instead of relying on signaling mechanisms (for example, RSVP), Diff-Serv relies on the Differentiated Service (DS) byte in the IP header.
- **IEEE 802.1p:** This IEEE standard specifies a priority scheme for Layer 2-switched LANs. As a Layer 2 scheme, IEEE 802.1p needs to be mapped to Diff-Serv as soon as the packet leaves a LAN domain.

As shown in Table 1, QOS requirements for IP Telephony can be established in several ways. Once the requirements are invoked, IP Telephony packets receive priority treatment by all switches and routers in the network. The edge switches provide traffic classification, identifying and "marking" the IP Telephony packets as highest priority. IP packets with similar requirements are handled at each network node by a series of queues. Queue management—including scheduling—governs how the various queues are serviced across the available bandwidth.

Vendors are embracing these standards for their routers and switches, controlled under a unified structure for policy, service and network management. The QOS and traffic-management capabilities required for deterministic and predictable treatment of voice traffic are being implemented in both WAN and LAN packet-based infrastructures.

There are multiple mechanisms for achieving the reliability required to support telephony applications. Those include robust switching platforms, multi-link trunking in campus networks, dynamic alternate routing and network management tools. Telephony-grade transmission quality is also provided by multi-homing inter-switch trunks, distributing traffic across multiple modules at every switch and routing switch. In addition, there is the ability for a call to be switched to the public network—transparently to the caller—if voice quality over the IP network degrades.

TABLE 1: Establishing QOS Requirements for IP Telephony

Layer	Mechanism
Layer 1	Physical Port
Layer 2	IEEE802.1p Priority Bits
Layer 3	■ IP address ■ Explicit session signaling —e.g. RSVP ■ Explicit packet fields using DS bits in the packet header
Layer 4	Examination of the protocol or socket ID

Enterprise IP Telephony: Payoffs in the WAN

The first round of enterprise IP Telephony solutions unify data and telephony traffic on a wide-area network. The payback period is between six and 18 months, depending on the networking environment. IP Telephony is one of three WAN unification options for enterprise users, along with ATM and frame relay.

Nortel Networks has been a pioneer in developing packet voice technologies, and today has more than two million packetized voice lines operating in enterprise networks, more than any other vendor.

Probe Research forecasts that by 2002, 18.5% of all U.S. domestic phone traffic will be carried over data lines—IP, frame relay and ATM.

There are two IP Telephony WAN unification configuration options. The first emulates a private-line configuration, mapping each configured PBX trunk onto a pre-configured IP "logical connection" (see Figure 1). This connection is defined by the source-destination address and logical port and is analogous to a permanent virtual circuit (PVC) with frame relay and ATM.

Speech activity detection and voice compression are used to achieve significant bandwidth efficiencies. Current implementations are adding IP QOS capabilities so, for example, IP trunk gateways can monitor IP network performance and dynamically adapt to network conditions. This enables the gateways to route voice calls over alternate routes (such as the PSTN) to avoid congested IP links.

The second configuration option emulates tandem PBX functionality by providing on-demand connectivity across a network (Figure 2) . An analogy in frame relay and ATM networks is switched virtual circuits (SVC). In this case, the IP network interprets PBX signaling on a call-by-call basis and converts the called number into a remote IP address. The challenge is to support the broad range of signaling systems around the world.

New Clients and Applications

The ability to achieve quick payback in the WAN has been the foundation for most business cases analyzing a migration to IP Telephony. Longer term, however, there is an even more significant justification: IP Telephony expands the number of applications that can run over a new switched, unified infrastructure (Figure 3).

Importantly, as Figure 3 shows, IP Telephony also broadens the number of end-user clients that can attach to an enterprise network. These include IP-enabled single- and multi-line phones, IP-enabled wireless sets, multi-function PDAs, Universal Serial Bus (USB)-based phones working with PCs and software clients running on the PC. Gateways will provide connectivity and feature interworking between these new devices and the traditional wireline and wireless voice devices.

IP Telephony solutions also facilitate porting applications onto open platforms, such as Windows NT and Unix. Unified messaging, for example, will achieve new levels of features and functionality with IP Telephony devices. Similarly, Web-enabled call centers will provide customer-service centers with greater transaction-management capabilities. Speech-activated applications and IVR applications are already playing significant roles. Mobility applications cater to telecommuters and mobile users, providing virtual desktop voice and data capabilities wherever they are.

A Prescription For Success

With IP Telephony still relatively new, there is no single set of implementation guidelines. However, there has been enough experience to create a framework for implementation, steps that enabled the early adopters to migrate successfully to IP Telephony, whether they were implementing a small, single site or multiple locations.

■ **1. Examine all your options for WAN consolidation:** Ascertain the opportunities and implications of unifying voice and data in the WAN. Not only can this positively impact cash flows, it establishes a WAN infrastructure with the highest price/performance ratio.

FIGURE 1 IP Telephony Emulating PBX Trunking

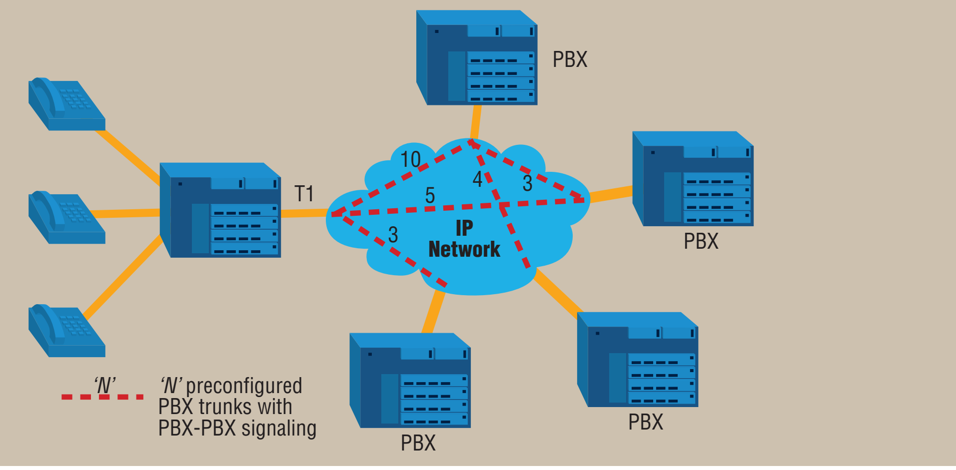


FIGURE 2 IP Telephony Emulating Tandem PBX Switching

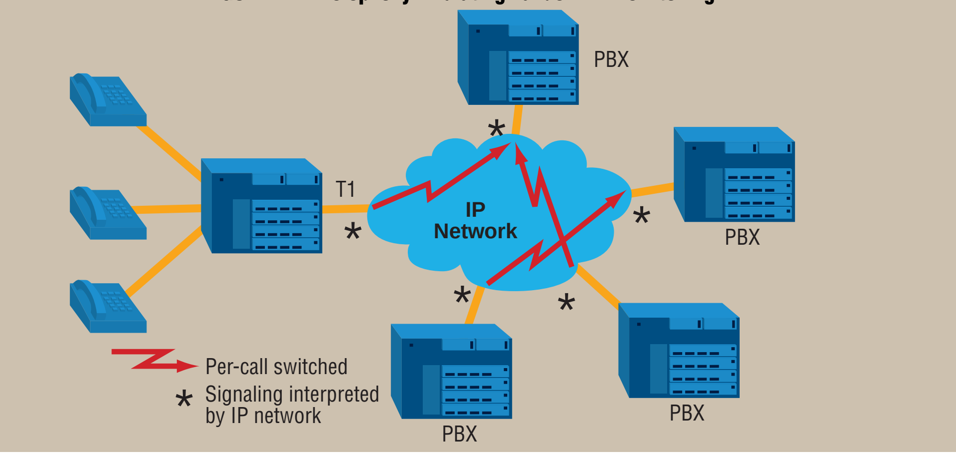
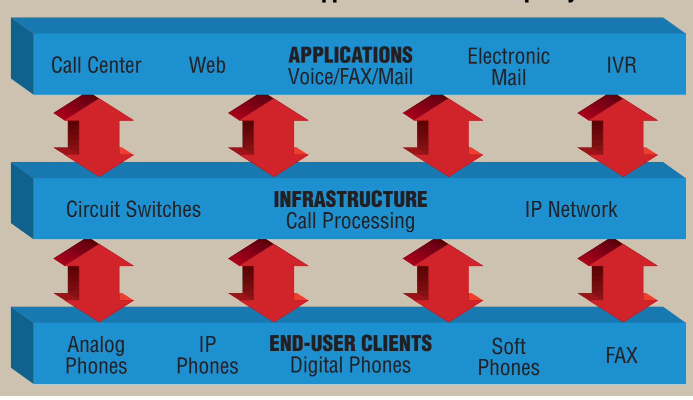


FIGURE 3 New Clients and Applications Via IP Telephony



The real payoff: IP Telephony's effect on applications

This article has focused on IP Telephony, but there are multiple technologies from which to choose: voice over IP, voice over frame relay and voice over ATM. Each has its own set of tradeoffs in terms of functionality and economics. Selecting one will involve an assessment of your application requirements, the readiness of your IP network to support telephony traffic and your infrastructure branch and backbone networks. For example, voice over IP may be preferred if multiple Layer 2 transport mechanisms are used and IP QOS has been deployed in the WAN.

■ **2. Invest in switched LAN architectures in the campus:** Shared-media approaches aren't adequate for IP Telephony. A two-tier, QOS-enabled architecture is required with Layer 2 switches in the wiring closet and Layer 3 routing switches in the campus backbone.

■ **3. Develop a QOS strategy before you deploy equipment:** You will need to roll out QOS handling, policy interfacing and "application awareness" at the edge of the network, for both IP Telephony and other business-critical applications such as SAP. Your QOS strategy must go beyond processing QOS requests to include mapping the requests into network-wide QOS mechanisms such as DiffServ.

Policy support consists of interfacing switching devices into the enterprise policy management system. Application awareness includes examining traffic at Layer 4 and even higher layers if the application cannot signal its QOS needs.

■ **4. Develop a system-level approach to enhancing data network reliability:** Telephony-grade reliability requires significant improvement to reliability of existing servers, which, on average, are down seven minutes per day. A system-level approach includes deploying highly robust switching platforms with network-level redundancy via multi-link routing and congestion-control mechanisms—all supported through comprehensive network and service management tools.

■ **5. Integrate voice and data communication under a single IT management organization:** It's easier to make the right business tradeoffs when voice and data planning and operations are under a single group.

IP Telephony—It's About Your Business Applications

IP Telephony means different things to different people. Much of the hype surrounds "free" calling via the Internet. And indeed, there are major economies to be gained by migrating to IP Telephony—total cost of ownership as well as short-term transport costs will fall.

But the real payoff is how IP Telephony and convergence ultimately affect your network applications. What's really important is installing capability that enriches person-to-person, real-time communication with multimedia capabilities.

Ultimately, the goal is to ensure that advanced, broadband multimedia communication becomes as reliable, simple and secure as voice communication is today. It's also about enhancing customer service for Web and ecommerce applications by bridging the telephony and data worlds for competitive advantage. Fundamentally, we're all in the business of bringing down barriers and empowering people via access to whatever information and expertise they need.

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IP Telephony—Building the Business Case

While company-specific metrics and issues drive the development of any business case, consider the following when considering migrating to IP telephony:

■ **Lower Costs for WAN Service:** Bandwidth is shared for multiple applications, and new, more sophisticated coding, silence suppression and compression schemes can maintain voice quality at a fraction of the bandwidth.

■ **Reduced Costs for Infrastructure:** Interconnecting end-user telephony clients over LAN infrastructures allows sharing of equipment between telephony and data networks, and helps reduce in-building and campus equipment and administration requirements.

■ **Simplified Routing Administration:** Integrate the voice network routing tables with distributed, dynamic IP routing systems such as OSPF.

■ **Mobility:** IP networks associate names with users rather than physical ports on a network. This makes it easier for telecommuters and road-warriors to receive calls and access telephony features from temporary locations. The flexibility of IP networks also helps simplify moves, adds and changes within telephony networks to help reduce costs.

■ **High-fidelity Voice:** Compression schemes can deliver toll-quality voice at much lower bandwidth, and higher sampling rates of the voice signal deliver higher-quality audio.

■ **Unified User Interface:** Make these easy to learn and use—for example, through intuitive browser-based graphical user interfaces.

■ **Application Integration:** Most businesses rely on telephony applications such as voice mail, IVR that provides text-to-speech conversion and application enablement based on speech or telephone key strokes. Similarly, CTI is found in call centers and enables specialized call handling and screen-pops. The unification of data and telephony networks enables integration to move to the next plateau—to the application and presentation layers. One example of the new level of integration is Web-enabled call centers and applications.

■ **Directory Integration:** IP Telephony enables integration of multiple directories into a cohesive system, thereby eliminating errors and labor caused by replication of user information.

■ **Policy Integration:** Telephony has long made possible specialized treatment of calls—for example, call routing based on user profiles and class of service, and calling privilege controls. Data networks also use policies to control access to information, manage network resources and provide differentiated treatment. IP Telephony enables integration of policies and rules.

■ **Network Management Integration:** Unifying the voice and data networking environments significantly reduces the total cost of ownership by integrating policy, service and network management capabilities.

Telephony: The "Killer App" for IP

Ravi Narayanan

During the past decade, the explosion in computing power and data traffic has led to dramatic advances in all areas of computing and networking. However, despite these changes, voice and data have remained largely separate disciplines. That distinction is now beginning to break down, and that in turn has profound implications.

For example:

- Voice traffic will be adapted and optimized to ride a data-optimized wide-area network.
- The dramatic improvements in Ethernet's price/performance make it possible to consolidate voice and data within campuses using Ethernet.
- Network architectures will migrate from the traditional "hub-and-spoke" to more distributed architectures, where traffic-management tasks such as packet forwarding and real-time traffic priority handling, and background control tasks such as call processing, are separated and distributed across the network.

Telephony over Data Networks

Telephony plays a central role in this network evolution; indeed, it is becoming the "killer app" of next-generation data networks. Telephony can be implemented across ATM-, frame relay- and IP-based data networks, but the degree to which each of these protocols can embrace telephony differs as shown in Table 1.

- **ATM:** Advances in voice over ATM in terms of quality of service and product availability will enable ATM to continue playing a major role in enterprise WAN backbones and service provider core infrastructure applications.
- **Frame Relay:** Because of the widespread availability of public, tariffed, frame relay services, and the advances in predictable transport of voice-over-frame relay, it will remain a favorite for enterprises to access voice services.
- **IP:** Its popularity in the WAN, LAN and applications will make IP the unifying protocol for telephony solutions. Among packet-switching technologies, IP is uniquely suited for this role because:

1. Many application-sensitive protocols, such as TCP, FTP, HTTP and RTP, are built around IP, as are the emerging protocols for packet-telephony, such as H.323 and MGCP.

2. IP is ubiquitous throughout LANs, campus networks, enterprise intranets and the Internet. Many user and application interfaces are evolving to leverage the IP-based browser, and IP addresses, names or URLs that have become universally recognized.

3. Scalability: Networks based on IP make applications distance-insensitive and location-independent, which means network expansion and evolution become simplified.

Impact of IP on Telephony

The primary impact of IP on telephony will be architectural—the functional components of traditional telephony will be distributed around IP networks, and much of the network intelligence will be provided by IP's dynamic networking capabilities. If this new architecture were to be built from scratch—in a "green field" environment, where no circuit-switched infrastructure exists—IP-based telephones would be distributed around an IP network, supported by a call server that provides call- and connection-management services.

However, in most of the world, circuit-switched infrastructures not only exist, they play an essential role. So the evolution to IP will take time, and IP-based telephony networks will coexist with traditional circuit-switched telephony networks.

This fundamental reality has significant implications for enterprise network managers: Users are going to require a lot more than just IP phones and simple call servers.

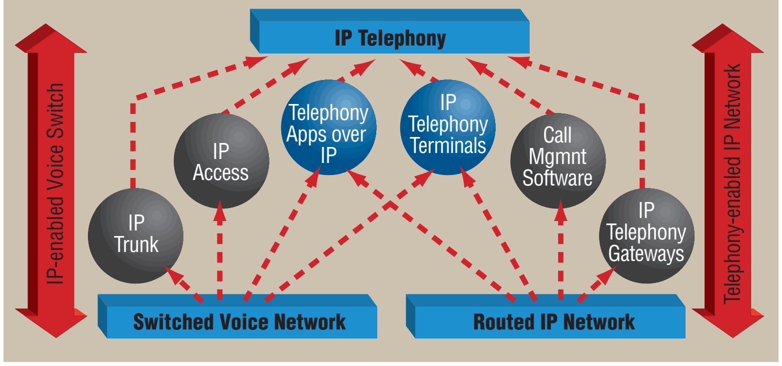
Enterprises will migrate to IP Telephony along one of two key migration paths: Either by transforming a circuit-switched telephony network to leverage IP, or by enhancing an IP network to provide telephony services and applications. Either way, a host of functional components will be required, but the emphasis will shift based on the starting point, as shown in Figure 1 (p. 8).

- **IP-enabled Voice Switches:** New enhancements to PBXs and key systems will become available to leverage IP networks while protecting existing investments. The enhancements include IP trunking,

TABLE 1 Packet Networks in Telephony Implementation

Telephony Capability	ATM	Frame Relay	IP	
Voice coding and packetization	■	■	■	WAN
Voice transport over WAN	■	■	■	WAN
Call management across the network (trunkside)	■	■	■	WAN
Voice transport over LAN			■	LAN
Packet-enabled voice terminals			■	LAN
Call management at lineside			■	LAN
Integrated presentation layer (user interface, etc.)			■	APPS
Integrated business and productivity applications			■	APPS
Integrated management (directory, policy, etc.)			■	APPS

FIGURE 1 Evolution to IP Telephony



IP-enabled telephony clients and access to telephony applications over IP.

- **Telephony-enabled IP networks:** Telephony will become a network-wide application of IP networks, WANs and LANs. New solutions will include voice-over-IP gateways that cover a range of price, performance and deployment requirements; IP-enabled telephony clients optimized for different applications such as high-fidelity, packet-based phones and mobility; distributed call management services, and applications such as Internet-based call centers and the integration of Web- and voice-processing that will enable speech-based surfing and text-voice conversion.

Note that these two potential starting points for migrating to IP Telephony aren't mutually exclusive. Indeed, many enterprises will pursue both because they have infrastructures for both telephony and IP already in place.

IP Telephony Solutions

There are four broad functional groups of IP Telephony solutions, as shown in Figure 2.

1. IP Telephony gateways: Gateways provide translation between circuit-switched voice networks and IP networks. There are different types, designed to match various price/performance and deployment requirements.

- IP-enabled PBXs and key systems: These gateways, tightly integrated with PBXs and key systems, deliver capabilities like call routing, trunk selection and telephony class-of-service. They deliver the reliability and scalability of traditional voice systems.
- Telephony-enabled routers, access devices (FRADs) and enterprise network switches: These gateways are closely coupled with traditional routing, bandwidth management and traffic management capabilities, and share the same levels of reliability and scalability.
- Standalone gateways: Translation devices whose sole purpose is to provide the gateway functions.
- Analog station gateways: These adapt analog devices, such as phones, fax machines and modems, to communicate through IP networks.

2. IP-telephony terminals: These end-user desktop devices include:

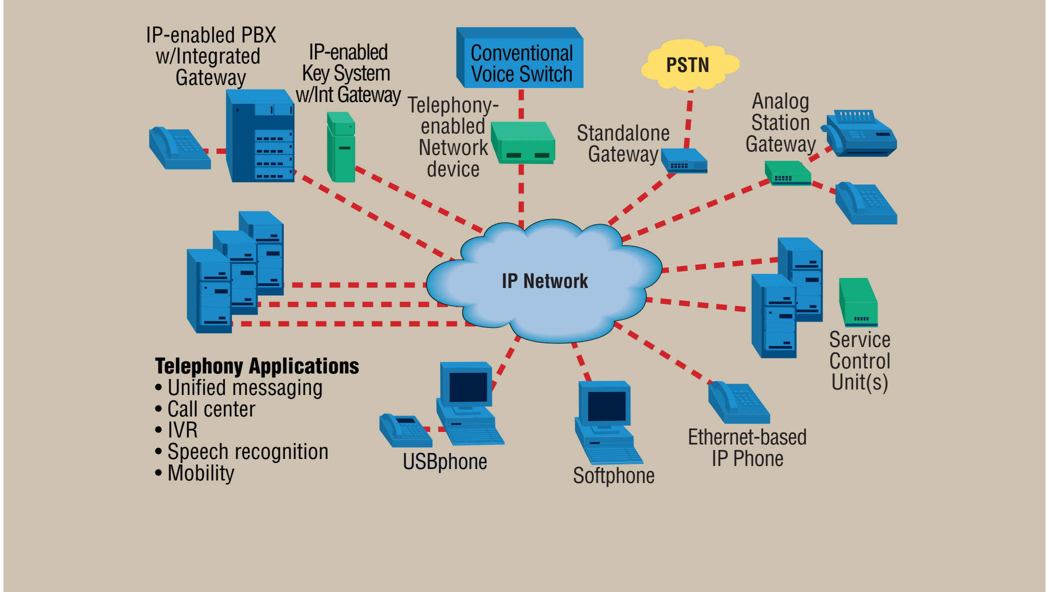
- IP-enabled phones.
- Softphones: PC-based software, which enables multimedia communication and telecommuting.
- Universal Serial Bus (USB)-based phones, which work with PCs and enable remote users and telecommuters to use one desktop device, supported by a single wiring plant, and still have access to the familiar telephone user interface.

3. Telephony applications: Familiar applications, which provide value-added capabilities in the context of converged networking, and include:

- Unified messaging applications that integrate voice mail, email and fax mail.
- Call-center applications that route and process calls to the appropriate person or agent based on specific skill sets—for example, language skills and/or technical knowledge.
- Interactive voice response (IVR) and speech-activated applications, which use telephony tones and human voice to recognize user inputs, and which provide text-to-voice conversion.
- Mobility applications so telecommuters and traveling workers can easily receive and make calls.
- Collaborative applications that enable geographically-dispersed employees to work together more productively.

4. Service Control Units (SCUs): These are specialized processors that combine the call-processing functionality of a PBX (or a key system)

FIGURE 2 IP Telephony Functional Components



and the "gatekeeper" functionality necessary to manage IP Telephony resources such as gateways, clients and network capacity. The functions performed include coordinating address translation (between telephony and IP address spaces), call signal processing, call establishment and related management, resource management and admissions control into an IP network environment.

An SCU provides the application-programming interface (API) to integrate telephony applications into an IP-telephony environment, and can be replicated for high availability and networked for scalability. Two types of enterprise IP Telephony SCUs will become prevalent over time, each of which will interwork with all the others:

- **SCUs that are hardware independent and server based:** Software components implemented on standard platforms, such as Windows NT servers, to enable IP networks to offer telephony as a service.
- **IP-enabled PBXs and key systems:** Call-processing intelligence integrated with IP Telephony call control to integrate trunk gateways and IP Telephony clients.

Quality and Reliability

The promise of IP Telephony won't be fulfilled until IP networks can deliver the Quality of Service (QOS) and reliability that telephone networks have been providing for decades. The elements of the infrastructure that will be effected are edge switches (for desktop connectivity), core routing switches (for campus backbone and server connectivity), and WAN switches and routers (for WAN access and connectivity).

The next-generation LAN and WAN switches, routers and routing switches will offer telephony-grade reliability through:

- Redundancy: dual power supplies, fan trays and support for hot-swap of critical components.
- Dual-active switch fabrics, to share traffic and provide non-disruptive recovery from hardware or software failure.
- Resilience at Layers 1, 2 and 3 of the ISO stack to cope with failures of hardware, inter-switch trunks and routing functions.
- Multi-homing inter-switch trunks, distributing traffic across multiple modules at every switch and routing switch.

IP Telephony solutions must be carefully architected

Key Protocols for IP Telephony

The goals of any effective IP Telephony solution must include:

- To enable speedy, cost-effective implementation of gateways, terminals and applications.
- To simplify and thus attain higher network reliability in gateways and terminals.
- To use standard computing platforms for efficient and speedy development of IP Telephony functionality.
- To be able to scale from a handful to thousands of gateways, terminals and appliances.

To satisfy these requirements, complex processing and background functions will be implemented in the Service Control Units (SCUs) described in this article. IP Telephony architectures comprising SCUs, gateways and terminals will use different groups of protocols, the two most prominent being H.323 and MGCP.

- **H.323:** This is the ITU-T standard for IP Telephony call control. It has the following characteristics:
- Treats gateways and terminals (clients) alike.
- Employs a peer-to-peer model in which the source terminal and/or gateway is a peer of the destination terminal and/or gateway.
- Optionally requires a gatekeeper function to provide telephone number-to-IP address translation, zone admission control and other resource-management functions.
- Requires gateways and terminals to provide their own call control/processing functions.
- Provides multiple options for voice, data and video communication.

H.323 is most applicable for implementation in end-points that have integrated processing power. These include PC-based IP Telephony clients and VOIP gateways integrated with PBX and key systems with inherent call-processing power. H.323 is the most widely employed standard among first-generation VOIP solutions.

- **Media Control Gateway Protocol (MGCP):** This IETF initiative for IP Telephony device and gateway control results from the merger of two other independent efforts: Bellcore's Simple Gateway Control Protocol (SGCP) and IP Device Control (IPDC).

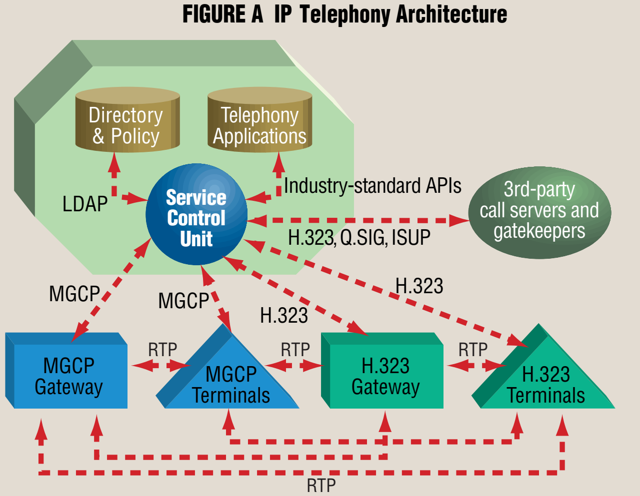
MGCP is expected to include extensions to support simple IP Telephony terminals, and it has the following characteristics:

- Originally intended for gateways but can be adopted for terminals (clients).
- Call-processing function is decoupled from gateways and terminals, and integrated with the gatekeeper function in a central service control unit.
- Optimized for telephony, as opposed to multimedia.

MGCP requires limited processing intelligence for IP Telephony operations at end-points such as gateways and terminals, which means the protocol simplifies and lowers the cost of implementation at the end points. MGCP is suitable in gateways that don't have call-processing intelligence, such as routers optimized for packet-forwarding and analog station gateways.

IP Telephony architectures will evolve to include the two protocols in different scenarios, supported by service control units, as shown in Figure A. Standard networking protocols such as ISDN will be implemented for network-level interworking between voice domains.

FIGURE A IP Telephony Architecture



Edge switches will provide traffic classification using Internet Engineering Task Force (IETF) Differentiated Services. This will enable edge switches to identify and "mark" IP Telephony packets as highest priority.

Traffic classification at the edge also will eliminate the performance impact of CPU-intensive QOS processing on core switches and routers, which have become a major network bottleneck. As a result, the future core routing switches will handle higher volumes of inter-subnet traffic (routing) with less processing requirements, and provide more granular traffic classification and queuing. This in turn will enable reliable and timely delivery of IP Telephony across the campus core into the WAN.

The bottom line: Routers and routing switches in the middle of the network will leverage the processing power of edge-based LAN switches; much of the heavy, processing-intensive work will be complete before traffic ever hits the routers.

U.S. Navy Uses Ship-To-Shore IP Telephony

- Conducting real-time, multipoint, multimedia ship-to-shore communications.
- Passing voice and data packets successfully up to 70 nautical miles out to sea.
- Holding live debriefings.

In 1997, those capabilities were long-term goals for the Naval Warfare Assessment Station (NWAS) of the U.S. Navy. Today, they're realities, thanks to the V/IP Phone/Fax IP Gateway from Nortel Networks, and the result has been significant cost savings, much quicker analysis of important information and a higher state of readiness.

"The V/IP Gateway is an integral component of the ship-to-shore communication network we created—named Line of Sight—for the Navy's AEGIS Combat System, the world's most advanced shipboard system for anti-aircraft and anti-missile defense on its destroyers and cruisers," said telecommunications engineer Matthew Layden.

"Older technology was replaced by a ship-to-shore wide-area network using IP over a spread-spectrum link," Layden added. "By digitizing, compressing and routing intra-organizational voice and fax traffic, the V/IP card enables the NWAS to simultaneously talk to engineers aboard ship, and share data with shipboard computers over the spread-spectrum link using IP."

Before V/IP, data had to be recorded onto nine-track tapes, and then transported via helicopter from the ship to the Atlantic Fleet Weapons Training Facility (AFWTF) at Roosevelt Roads naval base in Puerto Rico. There, engineers would use a Unix workstation to perform a preliminary data analysis, which was sent back the next day. In order to discuss the preliminary results, military officers and systems engineers had to resort to UHF radio—a stilted, one-way communication medium.

With the new implementation, V/IP-equipped Pentium PCs were deployed both on the ships and various base stations on land. On the ship, a V/IP-equipped PC sits between a single-line phone and a router, which sends calls to the radio frequency (RF) assembly and, from there, the signal is transmitted via an omni-directional antenna. On shore, a modified 10-foot telemetry antenna receives the transmission and routes it to another V/IP-equipped PC, which is connected to either a single-line phone or an audioconferencing bridge located on the secure side of the network.

Calls originating from the ship on the unsecured side of the network go to a PBX on shore. Data encryptors are on the ships and every shore node to protect sensitive and classified transmissions.

"With V/IP, NWAS has eliminated the need for a helicopter shuttle, which cost $38,000 per day," Layden said. "We also were able to prevent prolonging the number of days involved in Combat Ship System Qualification Tests (CSSQT), which cost $129,000 per day, not to mention travel costs.

"Even more importantly," Layden concluded, "the Navy can process the information gathered from the mock trial in a matter of hours—not months. Also, debriefs are held in a format which includes all shipboard personnel involved in the training—not just the officers but the actual sailors pushing the buttons and making things happen. This increases training value and our productivity tenfold."

Conclusion

The first-generation VOIP implementations consolidated voice- and call-processing functionality in devices like routers, which was adequate for deployments that aimed to prove that voice could be carried across IP networks with acceptable quality. As IP Telephony gains momentum, however, gateways and IP Telephony terminals will proliferate to meet a range of price/performance and deployment scenarios. Similarly, many telephony applications will evolve from the union of technologies for voice processing and the Internet.

This diversity highlights the need for IP Telephony solutions that are carefully architected. It makes no sense to simply transfer the complex telephony processing from one type of network concentration device—like a PBX—to another—like a router—without appreciable gains in price, performance or speedy evolution of applications.

Clearly, however, for the convergence of telephony and data networking to occur, an evolution in capabilities is needed on both sides. For example:

- Voice switches will evolve, becoming IP-enabled, enabling users with significant investments in voice switches to leverage the Internet and its related technologies, innovation and cost effectiveness.
- IP networks will evolve to offer the levels of quality and delay expected on telephone-grade networks. Similarly, IP-based phones, call management software and access to applications such as call centers will transform IP networks.
- Telephony and data networking will migrate from a hub-and-spoke architecture, where switching and routing devices provide device concentration, traffic management and call control, to a distributed architecture where these functions are distributed across a network to take advantage of cost-effective processing power and speedy feature and application deployment.

Over time, IP networks and the Internet will evolve from today's primary application—data and Web access—to provide a rich environment for people-to-people communications. Telephony, which has enabled humans to overcome the barriers of time and distance will then become the "killer app" of IP networks and the Internet.

Ravi Narayanan is the Director of Enterprise IP Telephony Strategy in Nortel Networks. Ravi has been instrumental in developing several breakthrough technologies including intelligent token-ring switching across WAN, VBR voice networking over ATM, and packet-based bandwidth managers that triggered the evolution of time-division multiplexing and circuit-switching to packet switching.

IP Telephony: Making the Business Case

Ann Swenson

It's kind of ironic: While new technology has helped businesses reach new levels of productivity and profitability, it's always difficult to convince CFOs to make a buy decision. Anyone making a case for new technology investments has to be able to answer questions such as: How is this going to improve our bottom line? Is it going to increase revenue?

Similarly, end users want to know: How will a new technology help me do my job better—or make my life easier?

IP Telephony isn't immune from these trends. Anyone pitching IP Telephony should expect to catch these same questions. Fortunately, there are some good answers.

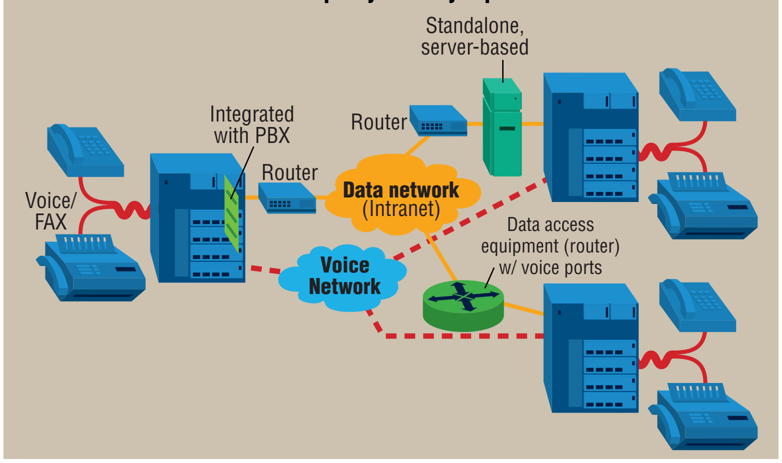
First, you need to recognize that there are multiple ways of implementing IP Telephony—voice-over-IP gateways, PC phones and LAN PBXs. The array of choices removes a major constraint: More than one migration path exists; each organization can pursue the path that makes the most sense to them.

Second, IP Telephony solutions address significant enterprise network goals: These include:

- Reducing costs.
- Consolidating communication facilities.
- Simplifying management and consolidating support staff.
- Improving customer service and increasing revenue.
- Improving productivity.
- Protecting existing investments.

Consider, for example, how IP Telephony gateways can affect the performance and cost of an enterprise network. Let's assume that your company has branch-office locations that need to be linked with a voice/data communication network.

FIGURE 1 IP Telephony Gateway Implementations



IP Telephony gateways can send real-time and non-real-time telephony information, such as voice, fax and voice mail, over a data network. The benefits include reduced cost through toll bypass—especially if one of the locations is outside North America—and/or consolidation of wide-area facilities. IP Telephony gateways also help preserve investments in networking systems, because existing PBX/key system phones, fax machines, routers and hubs usually remain in place.

Third, from a management/operations standpoint, IP Telephony gateway solutions offer:

- Simple administration and maintenance: Since an IP Telephony gateway ultimately enables voice and data to be carried on a single, unified network, network management will be streamlined.
- Transparent operation for users: Acting as a "translator" between the voice environment and the data network, most IP Telephony gateways enable use of existing desktop telephony devices. The process for making and receiving calls remains unchanged when an IP Telephony gateway is installed.

Implementation Options

There are multiple approaches to implementing IP Telephony gateways (Figure 1). These include:

- Standalone, server-based solutions.
- Adding voice ports to routers or LAN switches.
- Adding IP trunk cards to PBXs.

The first two options—external gateways such as router-based voice ports and external servers—make sense under specific circumstances. For example, if your current PBX does not support an integrated IP trunk card, the external approach is the only alternative. Similiarly, if your organization is planning a major upgrade to your data network, investige the external/server implementation models.

However, the key advantage to deploying integrated IP trunk cards into your PBXs is that there will be virtually no disruption/risk to operations; moreover, all of your existing investments in networking products are protected. Integrated IP Telephony gateway cards are administered and maintained like any other PBX trunk card, and basic features like call-detail records and class-of-service restrictions apply. Importantly, no special dialing is necessary—the service is transparent to users.

As an integrated component of the PBX, an IP trunk card delivers very high levels of reliability and availability. These cards can "look ahead" into the

network before setting up a call and thus ensure a high QOS (quality of service); calls can automatically be rerouted to traditional voice facilities when the data network cannot provide the required quality.

It's About More than Cheap Long Distance

The initial buzz about IP Telephony gateways was based on the ability to save on long-distance costs, via network consolidation or toll bypass. That benefit can be substantial, but there is more, much more to IP Telephony than cheap long distance rates.

For example, we're seeing innovative applications that marry the World Wide Web and call centers (Web-enabled call centers). Another important application of IP Telephony involves extending PBX functionality to remote workers via a single IP connection. Each of these is discussed below:

- **IP-enabled Call Centers:** In the same way that call centers have used voice mail and IVR (Interactive Voice Response) to improve customer service and increase revenues, call centers are increasingly looking for ways to enable transactions via the Internet and World Wide Web. We're already seeing specialized servers—called Email or Web-Request servers—that meet this emerging requirement.

  Customers can send an email requesting, for example, more information about your company's products, from your corporate website. That electronic message is routed to the most appropriate call-center agent for handling; this is analogous to skill-based routing familiar to voice-based call center applications. The server also monitors what's happened to minimize the risk of a customer not receiving a response.

  As the public Internet evolves to support high-quality, real-time, voice communications, rapid handling of Web-based customer interactions can be complemented with voice-button solutions: real-time voice interaction with the call-center agent. These solutions exist today, but tend to be problematic; quality of service for real-time voice over the Internet tends to be highly variable. A "call-back" scenario, where the customer provides a telephone number to the call center agent for a return call over the telephone network, promises a higher quality connection for voice.
- **Remote workers:** The ubiquity of IP network access can be leveraged to enable a telecommuter to extend his or her desktop—voice and data to remote sites, including home, airport or hotel room. Using a single dial-up connection, users can access data services—such as email, file and print services and database access—as well as important PBX-based services and features, such as voice mail, conference, hold and message-waiting.
- **Forwarding calls from the office to the remote worker's location:** The forwarding is transparent to the caller, who simply places a call using the employee's regular telephone number. For the remote worker, features like conference and transfer work just as if they were at the desk back at the office.
- **Placing and receiving calls over a multimedia computer:** In some applications, the personal computer essentially becomes an IP phone. IP Telephony software creates a "soft phone," with the handset usually replaced by a CTI headset. Alternatively, some applications support the use of a Universal Serial Bus (USB) phone which offers the benefit of a familiar interface—keypad and handset—for users.
- **Collaboration:** IP-based applications are emerging that significantly enhance real-time

Symantec—Pushing Convergence In The Here And Now

Everyone knows the line separating voice and data is blurring but, until recently, very few companies had figured out how to leverage the new reality. Symantec Corp., the world leader in utility software for business and personal computing, is studying how to make the most of voice/data integration and the promise of the Internet.

"The integration of voice and data isn't happening overnight," said Jason Conyard, director of worldwide communications for Symantec, "but it is happening, and we need to start understanding its implications and possibilities."

Nortel Networks' Meridian IP Telecommuter and Line Card will play an important role in Symantec's evolving strategy. "Convergence is about choice and about providing new applications," Conyard continued. "Meridian IP Telecommuter will give mobile employees more flexibility to access information and a chance to improve productivity."

The Meridian IP Telecommuter is Windows 95/98-based software client that turns a multimedia PC or laptop into a complete remote-office solution—IP data service as well as a real-time voice communication terminal—over a single analog line. Mobile users can call from homes, hotels, airports or other remote locations using Nortel Networks' Meridian 9617 Universal Serial Bus (USB) phone, which connects directly to a PC's USB port. There is an optional capability—using a headset with IP Telecommuter's softphone—which emulates common features of Meridian digital phones and gives users a more familiar interface.

Symantec initially implemented Meridian IP Telecommuter with six employees, and recently extended it to members of the management team in preparation for eventually rolling it out to a larger number of employees. "The trend is gradually moving from proprietary dial-up services for remote access, such as dedicated gateways for email, to more open systems," Conyard said. "We at Symantec want to provide a remote access portal not just for email but also for voice and video.

"Right now, I see us using Meridian IP Telecommuter for remote and mobile users to whom quality of service is important but not critical," he added. "As quality of service over IP matures, we can leverage IP more and extend those services to the desktop. Over time, I think we will see people use products like IP Telecommuter internally, as well."

One of the things Conyard likes about the product is that it offers the user a choice between using the softphone with a headset or a USB phone, which looks like a traditional handset. "People are very comfortable with handsets," he said. "No one is going to snuggle up on the couch or walk around the office with his or her PC.

"Nortel Networks is headed in the right direction with IP Telecommuter. It takes the concept of the mobile office to the next step," Conyard went on. "We envision it as a significant productivity enhancer. It is easy to set up, requires no extra hardware, will help reduce the cost of ownership, make network management simpler and increase integration of various capabilities at the desktop.

"We're really going to push Meridian IP Telecommuter and see what the challenges and opportunities are," he said. "People are starting to think of creative ways to use it. The product has turned the light on for people who've used it and made them realize that voice/data convergence is really coming."

communication and collaboration through the Web and existing telephony systems. They enable geographically-dispersed employees to create “virtual meeting rooms” on an intranet using voice, chat and document sharing. Logging into a virtual conference room is as easy as logging onto a Web page, and documents can be presented interactively to groups, accompanied by voice. These new applications represent the next step in the evolution of collaboration, moving to truly integrated multimedia, multi-person collaboration.

Success Factors for IP Telephony

A number of important considerations will affect whether a migration to voice/data convergence and IP Telephony succeeds. The product/technology-specific issues are discussed in the checklist on this page (“Checklist of Considerations”), while the company-specific considerations are discussed below.

■ **Public Internet vs. Managed Intranet:** The Internet is a collection of thousands of individual networks and organizations, and there is no centralized management. As new users gain access and as new services are offered, Internet traffic flow becomes less predictable and access becomes more difficult.

By contrast, private, IP-based data networks offer more manageability and control than the public Internet. With access and traffic flows regulated and controlled, network latency, delay and availability can be managed, and thus the network becomes more conducive to voice and other real-time traffic applications. While new standards and technologies will emerge to make the Internet acceptable for voice, for now, managed IP facilities are the only practical option.

■ **User acceptance:** IP Telephony solutions should retain familiar dialing plans and rely on intuitive end-user interfaces and/or familiar desktop appliances.

■ **Control of infrastructure:** Existing managed IP networks (private or virtual private networks) can be used to launch real-time, IP Telephony applications. However, it will be critical to assess and manage the effect telephony traffic will have on your data network. Some enterprise networks already have sufficient bandwidth and a design that is conducive to converged communications. But many others will have to phase in IP Telephony in conjunction with network upgrades in the areas of capacity and intelligence.

■ **Support organizations:** While some enterprises have been cross-training staff for years, in many others there are still barriers to unifying the technical support teams. However, make no mistake: These teams are necessary.

■ **Incremental implementation:** Anyone with experience in networking knows that flash-cuts to the unfamiliar rarely succeed. Migrating to converged networks and to IP Telephony on an incremental basis minimizes risks, lets you protect investments in existing equipment and provides a learning curve that is essential for your technical staff and end users.

■ **Reliability:** Does the system provide the level of reliability and uptime you require for your business operations? What are the backup capabilities in case of failure?

Conclusion

Can a business case be made for IP Telephony? Absolutely. The benefits are real—cost savings, simplified management and increased productivity and revenues—and the technology is available.

However, the key question is whether IP Telephony is right for your business. To determine the answer—and to identify the specific areas in which it would be most applicable—you must make a thorough evaluation of your enterprise network operations, the direction in which your business is moving and the appropriate networking tools that can help move it along the chosen direction.

Ann Swenson has product marketing responsibilities for Nortel Networks Meridian 1 IP Access and Applications, and she leads strategic market planning and market introduction activities for desktop and network IP Telephony applications in North America. Her experience also includes introducing ISDN PRI and BRI services for Meridian 1 communication systems.

There is much more to IP Telephony than cheap long-distance rates

Checklist of Considerations

When implementing IP Telephony solutions, enterprises must consider how well the proposed solutions deliver the following:

■ **Quality of Service:** How does the solution deal with concerns such as availability, latency, jitter, priority, predictability and delay? For characteristics such as delay, the IP Telephony equipment is only part of the equation; the performance characteristics of the data network must be taken into consideration as well. The actual network the voice will traverse can be affected by factors such as bandwidth and the number of router hops along the end-to-end communication path.

■ **Security:** Are PBX features such as class-of-service or authorization codes being leveraged?

■ **Accounting/billing:** Do you need all your call-detail records—are calls on your voice network and data network captured in a centralized manner?

■ **Management:** How will the solution be managed? Can it be incorporated with existing voice-management solutions? Is it easy and intuitive? Can it support SNMP alarm traps?

■ **Interoperability:** Is this a proprietary solution or one based on IP Telephony standards? Are your choices for voice compression based on industry standards?

■ **Scalability:** How many ports are needed for the amount of traffic expected over your IP network? Can ports be added simply, or is there a point at which potentially expensive additional hardware will be required?

■ **Cost—both short-term (product) and overall cost of ownership:** Product cost is only part of the cost equation. What is the total cost of ownership? Can the solution be managed and maintained by existing staff? Is maintenance covered in existing agreements? Are change-outs or updates required elsewhere in the network to support the solution being considered?

■ **Investment protection:** How well does the solution fit into the existing communications infrastructure? What is the vendor’s reputation for protecting investments? Can the solution be adapted to support additional capabilities in the future?